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EXAMINER	
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Please find below and/or attached an Office communication concerning this application or proceeding.

Office Action Summary	Application No.	Applicant(s)	
	09/432,917	MCDOWELL, SAMUEL KEITH	
	Examiner	Art Unit	
	Michael N. Opsasnick	2655	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If the period for reply specified above is less than thirty (30) days, a reply within the statutory minimum of thirty (30) days will be considered timely.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *02 November 1999*.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1-27 and 29-32* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1-27 and 29-32* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☒ Claim(s) 28 are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☒ The drawing(s) filed on *02 November 1999* is/are: a)☒ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)
2) ☒ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☒ Information Disclosure Statement(s) (PTO-1449 or PTO/SB/08) Paper No(s)/Mail Date 3.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .
5) ☐ Notice of Informal Patent Application (PTO-152)
6) ☐ Other: _____.

DETAILED ACTION

Election/Restrictions

1. Restriction to one of the following invention is required under 35 U.S.C. 121:

Group I, claims 1-27 and 29-32 drawn to a multi-channel interactive audio system, which is classified under class 704, subclass 500.

Group II, claim 28 drawn to a method for preparing PCM audio data for storage in a compressed format compatible with looping, which is classified under class 704, subclass 212.

Inventions I, claims 1-27 and 29-32 and II, claim 28 are related as combination and subcombination. Inventions in this relationship are distinct if it can be shown that (1) the combination as claimed does not require the particulars of the subcombination as claimed for patentability, and (2) that the subcombination has utility by itself or in other combinations (MPEP § 806.05(c)). In the instant case, the combination as claimed does not require the particulars of the subcombination as claimed because other grouping method exists, and invention 1 is used for audio signal mixing and compressing, while invention 2 can be used to create sound effects. The subcombination has separate utility such as sound effect generating systems.

During a telephone conversation with Mr. William Johnson on 1/5/2004 a provisional election was made with traverse to prosecute the invention 1, claims 1-27 and 29-32. Affirmation of this election must be made by applicant in replying to this Office action. Claim

28 is withdrawn from further consideration by the examiner, 37 CFR 1.142(b), as being drawn to a non-elected invention.

Claim Rejections - 35 USC § 103

2. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

3. Claims 1-4, 6, 10,11, 16-17, and 29-31 are rejected under 35 U.S.C. 103(a) as being unpatentable over Everett (US Patent No. 5,864,816) in view of ISO/IEC 11172-3 (Information Technology – Coding of moving pictures and associated audio for digital storage media at up to about 1,5 Mbits/s – Part 3: Audio, page 20-79, first edition 1993-08-01).

As per claims 1,29, Everett discloses a multi-channel interactive audio system (col. 4, ln. 39-40), comprising:

A memory for storing a plurality of audio components as sequences of input data frames, each input data frame including subband data and their scale factors that have been compressed and packed (as buffer22 of figure 2, containing frames of data with subband rows with scale factors – col. 4 lines 36-54));

A human input device (HID) for receiving input from a user (col. 5, ln. 46-60);

An application programming interface (API) that generates a list of audio components in response to the user input (col. 5, ln. 46-60, wherein an on-screen display shows data from various audio sources, fig. 3);

An audio renderer that

> Unpacks and decompresses the audio components' subband data and scale factors for each channel (col. 6, ln. 28-37);
>
> Calculates scale factors for the mixed subband data (as calculating scale factors to be applied on decompression -- col. 2 lines 13-15),
>
> Mixes the audio component's subband data in the subband domain for each channel (col. 4, ln. 45-54);

Everett fails to teach further compression of the mixed subband data and their scale factors for each channel, packing and multiplexing the channels' compressed subband data and scale factors into an output frame, and placing the output frame into a queue for transmission to a decoder. However, ISO/IEC 11172-3 teaches that an audio compression process wherein the mixed subband data and their scale factors for each channel (Coding of samples and bit-allocation sections on page 79), packs and multiplexes the channels' compressed subband data and scale factors into an output frame (Formatting and transmission section on page 79), and places the output frame into a queue for transmission to a decoder (Formatting and transmission section on page 79).

Therefore, it would have been obvious to one of ordinary skill in the art of MPEG coding systems to modify the teachings of Everett (US Patent No. 5,864,816) with the multiplexed output compression techniques of ISO/IEC 11172-3 because one of ordinary skill in the art

would readily realize that data compression at a centralized decoder before transmission to several remote systems in a limited bandwidth environment, would increase the transmission rate without compromising the quality of the signal (ISO/IEC 11172-3, page 74, first 11 lines)

As per claims 11 and 31, the combination of Everett (US Patent No. 5,864,816) in view of ISO/IEC 11172-3 teaches the audio render interfaces with an application that provides for sideways localization of the audio components, said audio renderer sideways localizing the audio components by applying a phase positioning filter to the subband data that spans the range from 200Hz to 1200Hz (under 2.4.3, the decoding process, subsection 2.4.3.4.10, teaching a polyphase filterband during decoding, wherein ISO/IEC dictates subbands (subsection 2.4.3.3.4), with frequency ranges between 200-1200 hz, page 66, subbsection C.1.3. → wherein the frequency subbands contain the range of 200-1200 hz).

As per claim 17, Everett (US Patent No. 5,864,816) discloses a multi-channel interactive audio system, comprising:

A memory for storing a plurality of audio components as sequences of input data frames in a bit stream that is coded with fixed length codes (FLCs) (Buffer22 of figure 2), each input data frame including a header, a bit allocation table, subband data and their scale factors that have been compressed and packed (figure 1);

A human input device (HID) for receiving input from a user (col. 5, ln. 46-60);

An application programming interface (API) that generates a list of audio components in response to the user input (col. 5, ln. 46-60, software program);

An audio renderer, which is hardcoded for the fixed header and bit allocation table (LUT element 36 of figure 2), that

Unpacks and decompresses the audio components' subband data and scale factors for each channel (col. 6, ln. 28-37);

Calculates scale factors for the mixed subband data (col. 3, ln. 58-65),

Uses the scale factors to determine the audible subband data (col. 3, ln. 54-57),

Mixes the audio component's subband data in the subband domain for each channel (col. 4, ln. 45-54);

Everett fails to teach further compression of the mixed subband data and their scale factors for each channel, packing and multiplexing the channels' compressed subband data and scale factors into an output frame, and placing the output frame into a queue for transmission to a decoder. However, ISO/IEC 11172-3 teaches that an audio compression process wherein the mixed subband data and their scale factors for each channel (Coding of samples and bit-allocation sections on page 79), packs and multiplexes the channels' compressed subband data and scale factors into an output frame (Formatting and transmission section on page 79), and places the output frame into a queue for transmission to a decoder (Formatting and transmission section on page 79).

Therefore, it would have been obvious to one of ordinary skill in the art of MPEG coding systems to modify the teachings of Everett (US Patent No. 5,864,816) with the multiplexed output compression techniques of ISO/IEC 11172-3 because one of ordinary skill in the art would readily realize that data compression at a centralized decoder before transmission to

several remote systems in a limited bandwidth environment, would increase the transmission rate without compromising the quality of the signal (ISO/IEC 11172-3, page 74, first 11 lines)

As per claim 2, Everett further discloses that the audio renderer mixes only the subband data that is considered audible to the user (col. 3, ln. 54-57).

As per claim 3, Everett further discloses that an audio renderer determines which subbands are audible to the user by using the listed audio components' scale factors to calculate the intra-subband masking effects and discard the inaudible audio components for each subband (col. 4, ln. 57-64).

As per claims 4,30, Everett teaches unpacking and decompressing on the subband (col. 3, ln. 54 to col. 4, ln. 67, more specifically col. 4 lines 60-64).

As per claim 6, Everett further discloses that the input data frame further includes a header and a bit allocation table that are fixed from frame-to-frame (figure 1) so that only the scale factors and subband data vary (col. 4, ln. 10-17).

As per claim 10, Everett further discloses that an audio render interfaces with an application that provides equalization of the audio components, said audio renderer equalizing each said audio component by modifying its scale factors (col. 3, ln. 58-65 or col. 4, ln. 60-62).

As per claim 16, Everett further discloses that one or more of the audio components comprise looped data having commencing input frames and closing input frames whose subband data has been preprocessed to ensure seamless concatenation with the commencing frame (col. 6, ln. 6-11).

4. Claim 5 is rejected under 35 U.S.C. 103(a) as being unpatentable over Everett (US Patent No. 5,864,816) in view of ISO/IEC 11172-3 (Information Technology – Coding of moving pictures and associated audio for digital storage media at up to about 1,5 Mbits/s – Part 3: Audio, page 20-79, first edition 1993-08-01), and further in view of Davis et al (US Patent No. 5909664).

As per claim 5, Everett (US Patent No. 5,864,816) further disclose that the audio renderer

a. stores the unpacked and decompressed subband data (Buffer 22 of figure 2);

b. for each subband, multiplies the audible subband data by their respective scale factor and adds them together into a sum (col. 5, ln. 1-9);

c. for each subband, multiplies the sum by the reciprocal of the maximum scale factor for the audible subband data to produce the mixed subband data (col. 3, ln. 61-65);

d. if the mixed subband data overflows the format, increments the maximum scale factor to the next largest value and repeats step c (col. 5, ln. 14-18).

Everett (US Patent No. 5,864,816) fails to specifically disclose a method for storing the unpacked and decompressed subband data in a left shifted format in the memory in which the sign bit of the N-bit subband data is aligned to the sign bit of the M-bit format and the M-N rightmost bits represent noise that is below a noise floor. However, ISO/IEC 11172-3 further teaches that the sign bit of the N-bit subband data is aligned to the sign bit of the M-bit format and the M-N rightmost bits represent noise that is below a noise floor (pages 70-71, where bspl or MNR represents for M-N, with MNR representing the mask to noise ratio. M is the frame size and N is a portion of the frame on page 75, then M-N represents the noise floor). Therefore, it would have been obvious to one of ordinary skill in the art of audio coding to modify the teachings of incorporate International Standard into Everett because the bit difference representing the mask-to-noise-ratio or noise parameter, is needed to provide feedback to enable the system to adjust control parameters to minimize errors (ISO/IEC 11172-3., page 74, lines 22-29.)

The combination of Everett (US Patent No. 5,864,816) in view of ISO/IEC 11172-3 still fails to disclose that a method for storing the unpacked and decompressed subband data in a left shifted format in the memory. However, Davis (US Patent No. 5909664) teaches that subband data is left shifted and the leftmost bit is the sign bit (col. 13, ln. 35-65). Therefore, it would have been obvious to incorporate Davis et al. into the combination of Everett (US Patent No. 5,864,816) in view of ISO/IEC 11172-3 because one of ordinary skill in the art, at the time of invention, would readily realize that storing data in a left shift format would enable the system to keep track of the position and sign of each data bits (Davis et al, col. 13 lines 35-41).

5. Claims 7-9 and 18 are rejected under 35 U.S.C. 103(a) as being unpatentable over Everett (US Patent No. 5,864,816) in view of ISO/IEC 11172-3 (Information Technology – Coding of moving pictures and associated audio for digital storage media at up to about 1,5 Mbits/s – Part 3: Audio, page 20-79, first edition 1993-08-01), and further in view of Tsutsui et al. (US Patent No. 6314391).

As per claim 7, the combination of Everett (US Patent No. 5,864,816) in view of ISO/IEC 11172-3 fails to specifically disclose that the compressed subband data is coded with fixed length codes. However, Tsutsui et al. (US Patent No. 6314391) teaches that the compressed subband data is coded with fixed length codes (col. 20, ln. 52-67). Therefore, it would have been obvious to one of ordinary skill in the art of data coding to modify the combination of Everett (US Patent No. 5,864,816) in view of ISO/IEC 11172-3 with fixed length codes because it would advantageously reduce processing time and power to make the system more efficient (Tsutsui et al. (US Patent No. 6314391), col. 20 lines 35-44).

As per claims 8 and 18, Everett further discloses a method for unpacking each piece of the N-bit subband data (col. 6, ln. 28-37), where N varies across the subbands (col. 4, ln. 16-17), as follows:

a. Utilizes the FLCs and fixed bit allocation to calculate the position of the subband data in the input audio frame (col. 3, ln. 40-42), extract the subband data and stores it in the memory as M-bit words where the leftmost bit is a sign bit (col. 4, ln. 47-51).

Everett fails to specifically disclose a method for (b) left shifting the subband data until its sign bit is aligned with the M-bit word's sign bit, the rightmost M-N bits remaining in the M-bit word as noise. However, International Standard further teaches the rightmost M-N bits remaining in said M-bit word as noise (pages 70-71, where bspl or MNR represents for M-N, but MNR=SNR-SMR. M is the frame size and N is a portion of the frame on page 75, then M-N represents the noise floor). Therefore, it would have been obvious to one of ordinary skill in the art of audio coding to modify the teachings of incorporate International Standard into Everett because the bit difference representing the mask-to-noise-ratio or noise parameter, is needed to provide feedback to enable the system to adjust control parameters to minimize errors (ISO/IEC 11172-3., page 74, lines 22-29.)

As per claim 9, Everett further discloses that the audio renderer is hardcoded for the fixed header and bit allocation table so that the audio renderer only processes the scale factors and subband data to increase speed (36 of figure 2, subsystem 36 explain a ROM, which is needed to store the header and the bit allocation table).

6. Claims 13, 19, 26 are rejected under 35 U.S.C. 103(a) as being unpatentable over Everett (US Patent No. 5,864,816) in view of ISO/IEC 11172-3 (Information Technology – Coding of moving pictures and associated audio for digital storage media at up to about 1,5 Mbits/s – Part 3: Audio, page 20-79, first edition 1993-08-01), and further in view of Rayskiy (US Patent No. 6278387).

As per claims 13,19,26, Everett (US Patent No. 5,864,816) discloses a multi-channel interactive audio system, comprising:

A memory for storing a plurality of audio components as sequences of input data frames in a bit stream that is coded with fixed length codes (FLCs) (Buffer22 of figure 2), each input data frame including a header, a bit allocation table, subband data and their scale factors that have been compressed and packed (figure 1);

A human input device (HID) for receiving input from a user (col. 5, ln. 46-60);

An application programming interface (API) that generates a list of audio components in response to the user input (col. 5, ln. 46-60, software program);

An audio renderer, which is hardcoded for the fixed header and bit allocation table (LUT element 36 of figure 2), that

Unpacks and decompresses the audio components' subband data and scale factors for each channel (col. 6, ln. 28-37);

Calculates scale factors for the mixed subband data (col. 3, ln. 58-65),

Uses the scale factors to determine the audible subband data (col. 3, ln. 54-57),

Mixes the audio component's subband data in the subband domain for each channel (col. 4, ln. 45-54);

Everett fails to teach further compression of the mixed subband data and their scale factors for each channel, packing and multiplexing the channels' compressed subband data and scale factors into an output frame, and placing the output frame into a queue for transmission to a decoder. However, ISO/IEC 11172-3 teaches that an audio compression process wherein the

mixed subband data and their scale factors for each channel (Coding of samples and bit-allocation sections on page 79), packs and multiplexes the channels' compressed subband data and scale factors into an output frame (Formatting and transmission section on page 79), and places the output frame into a queue for transmission to a decoder (Formatting and transmission section on page 79). Therefore, it would have been obvious to one of ordinary skill in the art of MPEG coding systems to modify the teachings of Everett (US Patent No. 5,864,816) with the multiplexed output compression techniques of ISO/IEC 11172-3 because one of ordinary skill in the art would readily realize that data compression at a centralized decoder before transmission to several remote systems in a limited bandwidth environment, would increase the transmission rate without compromising the quality of the signal (ISO/IEC 11172-3, page 74, first 11 lines)

As per claims 13 and 19, Everett discloses that an audio renderer transmits a sequence of output frames that provide real-time interactive multi-channel audio with the same format as the multi-channel audio (figure 2), but fail to specifically disclose a Digital Surround Sound decoder that is capable of decoding multi-channel audio. However, Rayskiy teaches surround sound audio channels (col. 8, ln. 18-23, this suggests that a Digital Surround Sound decoder is needed to decode these surround sound multi-channel audio. The advantage of using the teaching of Rayskiy in the modified Everett is to enable playback of surround sound multichannel audio. Since the modified Everett and Rayskiy are analogous art because they are from the same field of endeavors, it would have been obvious to one of ordinary skill in the art at the time the invention was made to further modify Everett by incorporating the teaching of Rayskiy order to playback of surround sound multichannel audio, which would offer better sound quality.

As per claim 26, the modified Everett still fails to disclose a digital surround sound decoder that decodes the output frames to produce multi-channel audio, the output frames having the same format as existing prerecorded multi-channel digital audio. However, Rayskiy teaches surround sound audio channels (col. 8, ln. 18-23, this suggests that a Digital Surround Sound decoder is needed to decode these surround sound multi-channel audio.

Since the modified Everett and Rayskiy are analogous art because they are from the same field of endeavors, it would have been obvious to one of ordinary skill in the art at the time the invention was made to further modify Everett by incorporating the teaching of Rayskiy order to playback of surround sound multichannel audio.

Regarding claim 27, the modified Everett still fails to specifically disclose a digital decoder that decodes the bitstream into a multi-channel audio signal, and a single bandlimited connector that delivers the bit stream to the decoder. However, Rayskiy teaches a digital decoder that decodes the bitstream into a multi-channel audio signal (109 of figure 1), and a single bandlimited connector that delivers the bit stream to the decoder (117 of figure 1, every connector has a bandwidth limitation). Since the modified Everett and Rayskiy are analogous art because they are from the same field of endeavors, it would have been obvious to one of ordinary skill in the art at the time the invention was made to further modify Everett by incorporating the teaching of Rayskiy in order to reconstruct the audio signal for playback.

7. Claims 14-15 are rejected under 35 U.S.C. 103(a) as being unpatentable over Everett (US Patent No. 5,864,816) in view of ISO/IEC 11172-3 (Information Technology – Coding of moving pictures and associated audio for digital storage media at up to about 1,5 Mbits/s – Part 3: Audio, page 20-79, first edition 1993-08-01), further in view of Rayskiy (US Patent No. 6278387), and further in view of Crowder Sr. (US. Patent No. 4,546,212).

As per claims 14-15, the common elements to claims 1 and 13 are rejected under Everett (US Patent No. 5,864,816) in view of ISO/IEC 11172-3 (Information Technology – Coding of moving pictures and associated audio for digital storage media at up to about 1,5 Mbits/s – Part 3: Audio, page 20-79, first edition 1993-08-01) under the same rationale as presented above. Furthermore, Everett discloses that an audio renderer transmits, in real-time and in response to the user input, the output frames as a unified and compressed bitstream over the single connector (figure 2), but fails to specifically disclose that data is transmitted via a bandlimited connector to the Digital Surround Sound decoder, which decodes the bitstream into the interactive multi-channel audio. However, Rayskiy teaches surround sound multi-channel audio (col. 8, ln. 18-23, this suggests that a Digital Surround Sound decoder is needed to decode these surround sound multi-channel audio).

Since the modified Everett and Rayskiy are analogous art because they are from the same field of endeavors, it would have been obvious to one of ordinary skill in the art at the time the invention was made to further modify Everett by incorporating the teaching of Rayskiy order to playback of surround sound multichannel audio.

The combination of Everett (US Patent No. 5,864,816) in view of ISO/IEC 11172-3 in further view of Rayskiy (US Patent No. 6278387) still fails to specifically disclose that the connector is a bandlimited connector and the bandwidth of the decoded bitstream exceeds the bandwidth of the bandlimited connector.

However, Crowder Sr. (US. Patent No. 4,546,212) teaches a telephone connector whose bandwidth is approximately from 0-3 KHz (col. 6, ln. 6-7), which is much less than the bandwidth of the decoded audio bitstream as mention in Everett (col. 3, ln. 45-54). The advantage of using the teaching of Crowder Sr. in the modified Everett is to transmit data at a higher rate than could be supported by the connector in the decoded format.

Since the modified Everett and Crowder Sr. are analogous art because they are from the same field of endeavors, it would have been obvious to one of ordinary skill in the art at the time the invention was made to further modify Everett by incorporating the teaching of Crowder Sr. in order to transmit compressed data at a higher rate than could otherwise be supported by the connector.

8. Claims 12, 20-21, 23-25, and 32 are rejected under 35 U.S.C. 103(a) as being unpatentable over Everett (US Patent No. 5,864,816) in view of ISO/IEC 11172-3 (Information Technology – Coding of moving pictures and associated audio for digital storage media at up to about 1,5 Mbits/s – Part 3: Audio, page 20-79, first edition 1993-08-01), further in view of Paneth et al. (US Patent No. 5734678).

As per claims 12, 20, and 32, Everett further discloses that the input and output frames also include a header and a bit allocation table (figure 1), but fails to specifically disclose the audio renderer providing for the seamless generation of output frames to maintain decoder sync by

> (a) Placing a null output template in the queue that includes the header, the bit allocation table and subband data and scale factors that represent an inaudible signal; (b) If the next frame of mixed subband data and scale factors is ready, writing the mixed subband data and scale factors over the previous output frame and transmitting the output frame; and (c) If the next frame is not ready, transmitting the null output template.

However, Paneth et al. teaches (a) Placing a null output template in the queue that includes the header, the bit allocation table and subband data and scale factors that represent an inaudible signal; (b) If the next frame of mixed subband data and scale factors is ready, writing the mixed subband data and scale factors over the previous output frame and transmitting the output frame; and (c) If the next frame is not ready, transmitting the null output template (col. 49, ln. 15-24, a null template is permanently stored in the transmit buffer 110. If the frame is ready, that frame is written over the null transmit buffer and transmitted. If the next frame is not ready, null template is transmitted).

Since the modified Everett and Paneth et al. are analogous art because they are from the same field of endeavors, it would have been obvious to one of ordinary skill in the art at the time the invention was made to further modify Everett by incorporating the teaching of Paneth et al.

in order to avoid transmission of inaudible data frames to reduce processing time and transmitting power for the system.

As per claim 21, Everett discloses a multi-channel interactive audio system, comprising:

A memory for storing a plurality of audio components as sequences of input data frames (Buffer 22 of figure 2), each input data frame including a header, a bit allocation table, subband data and their scale factors that have been compressed and packed (figure 1);

A human input device (HID) for receiving input from a user (col. 5, ln. 46-60);

An application programming interface (API) that generates a list of audio components in response to the user input (col. 5, ln. 46-60, wherein an on-screen display shows data from various audio sources, fig. 3);

An audio renderer that generates a seamless sequence of output frames by

b. concurrently unpacking and decompressing the audio components' data and for each channel, mixing the audio components' data for each channel (col. 6, ln. 31-35), calculating scale factors for the mixed data (as calculating scale factors to be applied on decompression -- col. 2 lines 13-15).

Everett fails to teach further compression of the mixed subband data and their scale factors for each channel, packing and multiplexing the channels' compressed subband data and scale factors into an output frame, and placing the output frame into a queue for transmission to a decoder. However, ISO/IEC 11172-3 teaches that an audio compression process wherein the mixed subband data and their scale factors for each channel (Coding of samples and bit-allocation sections on page 79), packs and multiplexes the channels' compressed subband data

and scale factors into an output frame (Formatting and transmission section on page 79), and places the output frame into a queue for transmission to a decoder (Formatting and transmission section on page 79). Therefore, it would have been obvious to one of ordinary skill in the art of MPEG coding systems to modify the teachings of Everett (US Patent No. 5,864,816) with the multiplexed output compression techniques of ISO/IEC 11172-3 because one of ordinary skill in the art would readily realize that data compression at a centralized decoder before transmission to several remote systems in a limited bandwidth environment, would increase the transmission rate without compromising the quality of the signal (ISO/IEC 11172-3, page 74, first 11 lines).

The modified Everett still fails to specifically disclose that (a) Placing a null output template in the queue that includes the header, the bit allocation table and subband data and scale factors that represent an inaudible signal; (b) If the next frame of mixed subband data and scale factors is ready, writing the mixed subband data and scale factors over the previous output frame and transmitting the output frame; and (c) If the next frame is not ready, transmitting the null output template.

However, Paneth et al. teaches (a) Placing a null output template in the queue that includes the header, the bit allocation table and subband data and scale factors that represent an inaudible signal; (b) If the next frame of mixed subband data and scale factors is ready, writing the mixed subband data and scale factors over the previous output frame and transmitting the output frame; and (c) If the next frame is not ready, transmitting the null output template (col. 49, ln. 15-24, a null template is permanently stored in the transmit buffer 110. If the frame is ready,

that frame is written over the null transmit buffer and transmitted. If the next frame is not ready, null template is transmitted).

Since the modified Everett and Paneth et al. are analogous art because they are from the same field of endeavors, it would have been obvious to one of ordinary skill in the art at the time the invention was made to further modify Everett by incorporating the teaching of Paneth et al. in order to avoid transmission of inaudible data frames to reduce processing time and transmitting power for the system.

As per claim 23, Everett further discloses that the audio data comprises subband data and its scale factors (figure 1), the audio renderer mixes only the subband data that is considered audible to the user (col. 3, ln. 54-57).

As per claim 24, Everett further discloses that an audio renderer determines which subbands are audible to the user by using the listed audio components' scale factors to calculate the intra-subband masking effects and discard the inaudible audio components for each subband (col. 4, ln. 57-64).

As per claim 25, Everett further discloses that the audio renderer unpacks and decompresses the audio data to determine the audible subbands (col. 3, ln. 54 to col. 4, ln. 67), but fails to specifically disclose that unpacks and decompresses the audio components' scale factors first, determines audible subbands, and then unpacks and decompresses only the subband data in the audible subbands. However, it would have been obvious to one of ordinary skill in

the art at the time the invention was made to decode only audible subbands by using the scale factor information in order to reduce processing time.

9. Claim 22 is rejected under 35 U.S.C. 103(a) as being unpatentable over Everett (US Patent No. 5,864,816) in view of International Standard (Information Technology – Coding of moving pictures and associated audio for digital storage media at up to about 1,5 Mbits/s – Part 3: Audio, page 20-79, first edition 1993-08-01), further in view of Paneth et al. (US Patent No. 5734678), and further in view of Rayskiy (US Patent No. 6278387).

As per claim 22, the modified Everett fails to disclose that the decoder is a digital surround sound decoder that is capable of decoding multi-channel audio. However, Rayskiy teaches surround sound audio channels (col. 8, ln. 18-23, this suggests that a Digital Surround Sound decoder is needed to decode these surround sound multi-channel audio. The advantage of using the teaching of Rayskiy in the modified Everett is to enable playback of surround sound multichannel audio so as to provide improved sound quality.

Since the modified Everett and Rayskiy are analogous art because they are from the same field of endeavors, it would have been obvious to one of ordinary skill in the art at the time the invention was made to further modify Everett by incorporating the teaching of Rayskiy order to playback of surround sound multichannel audio.

Conclusion

10. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure. Please see related art listed on the PTO-892 form.

11. **Any response to this action should be mailed to:**

Commissioner of Patents and Trademarks
Washington, D.C. 20231
or faxed to:
(703) 872 9314,
(for informal or draft communications, please label "PROPOSED" or "DRAFT")

Hand-delivered responses should be brought to Crystal Park II, 2121 Crystal Drive, Arlington. VA., Sixth Floor (Receptionist).

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Michael Opsasnick, telephone number (703)305-4089, who is available Tuesday-Thursday, 9AM-4PM.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Ms. Doris To, can be reached at (703)305-4827. The facsimile phone number for this group is (703)872-9314.

Any inquiry of a general nature or relating to the status of this application or proceeding should be directed to the Group 2600 receptionist whose telephone number is (703) 305-4750, the 2600 Customer Service telephone number is (703) 306-0377.

mno
4/17/2004

DORIS H. TO
SUPERVISORY PATENT EXAMINER
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